CLAIMS

1. A dual rate echo canceller for simultaneous support of a plurality of annexes, the echo canceller comprising:

an annex selector for selecting at least one of a plurality of annexes;

an echo cancellation filter having an input adapted to receive a transmit signal, the echo cancellation filter being adapted to generate an output signal comprising a signal component representative of an echo signal associated with the transmit signal, wherein the echo cancellation filter operates at a transmit rate; and

an output up sampling block having an input adapted to receive the output signal, the output up sampling block being adapted to generate an up-sampled signal by a factor associated with a selected annex.

2. The echo canceller of claim 1, wherein the up-sampled signal is generated by a zero filling operation.

3. The echo canceller of claim 1, wherein the echo cancellation filter is an adaptive finite impulse response filter.

4. The echo canceller of claim 1, further comprising an interpolation filter having an input adapted to receive the up-sampled signal, the interpolation filter being adapted to generate a first filtered output at a receive rate, wherein the first filtered output is subtracted from an incoming signal to generate a residual echo signal.

5. The echo canceller of claim 4, wherein the interpolation filter is a fixed or programmable low pass finite impulse response filter for interpolating the up-sampled signal.

6. The echo canceller of claim 4, further comprising an anti-aliasing filter having an input adapted to receive the residual echo signal, the anti-aliasing filter adapted to generate a second filtered output at a receive rate.

7. The echo canceller of claim 6, wherein the anti-aliasing filter is a fixed or programmable low pass finite impulse response filter for filtering out frequency signals above a transmit band.

8. The echo canceller of claim 1, further comprising a down sampling block having an input adapted to receive the second filtered output, the down sampling block being adapted to generate a down sampled output signal at a transmit rate by a factor associated with a selected annex.

9. The echo canceller of claim 1, further comprising a delay block having an input adapted to receive an input transmit signal, the delay block being adapted to generate a delayed transmit signal for compensating for a delay in the echo signal.

10. The echo canceller of claim 1, further comprising an up sampling block having an input adapted to receive an input transmit signal, the up sampling block being adapted to generate an up-sampled signal by a factor associated with one of a plurality of factors for a plurality of annexes, wherein an output of the input up-sampled block is coupled to an input of the echo cancellation filter.

11. The echo canceller of claim 1, wherein the plurality of annexes comprise Annex A and Annex B of G.992.1.

12. The echo canceller of claim 8, wherein the down sampled output signal comprises an error signal for adaptively at least one training coefficient of the echo cancellation filter.

13. The echo canceller of claim 12, wherein least mean square update rules are used to adaptively train the at least one coefficient of the echo canceller filter.

14. A method for implementing a dual rate echo canceller for simultaneous support of a plurality of annexes, the method comprising the steps of:

selecting at least one of a plurality of annexes;

receiving a transmit signal;

generating an output signal comprising a signal component representative of an echo signal associated with the transmit signal at a transmit rate;

receiving the output signal; and

generating an up-sampled signal by a factor associated with a selected annex.

15. The method of claim 14, wherein the up-sampled signal is generated by a zero filling operation.

16. The method of claim 14, wherein the echo cancellation filter is an adaptive finite impulse response filter.

17. The method of claim 14, further comprising the steps of:

receiving the up-sampled signal; and

generating a first filtered output at a receive rate, wherein the first filtered output is subtracted from an incoming signal to generate a residual echo signal.

18. The method of claim 17, further comprising the step of:

implementing a fixed or programmable low pass finite impulse response filter for interpolating the up-sampled signal.

19. The method of claim 17, further comprising the steps of:

receiving the residual echo signal; and

generating a second filtered output at a receive rate.

20. The method of claim 19, further comprising the step of:

implementing a fixed or programmable low pass finite impulse response filter for filtering out frequency signals above a transmit band.

21. The method of claim 14, further comprising the steps of:

receiving the second filtered output; and

generating a down sampled output signal at a transmit rate by a factor associated with a selected annex.

22. The method of claim 14, further comprising the steps of:

receiving an input transmit signal; and

generating a delayed transmit signal for compensating for a delay in the echo signal.

23. The method of claim 14, further comprising the steps of:

receiving an input transmit signal; and

generating an up-sampled signal by a factor associated with one of a plurality of factors for a plurality of annexes, wherein an output of the input up-sampled block is coupled to an input of an echo cancellation filter.

24. The method of claim 14, wherein the plurality of annexes comprise Annex A and Annex B of G.992.1.

25. The method of claim 21, wherein the down sampled output signal comprises an error signal for adaptively at least one training coefficient of an echo cancellation filter.

26. The method of claim 25, wherein least mean square update rules are used to adaptively train the at least one coefficient of the echo canceller filter.

27. The echo canceller of claim 1, wherein a least mean square update rule is applied to train at least one coefficient of the echo canceller filter, the least means square update rule is defined as

$$\mathbf{w}_{n+1} = \mathbf{w}_n - \mu e(n)\mathbf{X}_n^T\mathbf{h},$$

where $\mathbf{w}$ represents a coefficient vector, μ represents step size, $e(n)$ represents the error signal, and $\mathbf{X}_n^T\mathbf{h}$ represents a vector-matrix product where $\mathbf{X}$ is a Hankel matrix and $\mathbf{h}$ represents a filter coefficients vector.

28. The echo canceller of claim 27, wherein

$$e(n) = y_r(n) - \mathbf{h}^T\mathbf{X}_n\mathbf{w},$$

where $y_r(n)$ is a received signal component.

29. The echo canceller of claim 1, wherein a least mean square update rule is applied to train at least one coefficient of the echo canceller filter, the least means square update rule is defined as

$$\mathbf{w}_{n+1} = \mathbf{w}_n - \mu e(n)\mathbf{x}_{n-d},$$

where $\mathbf{w}$ represents a coefficient vector, μ represents step size, $e(n)$ represents the error signal, and $\mathbf{x}$ represents a data vector.

30. The echo canceller of claim 29, wherein

$$e(n) = y_r(n) - \mathbf{h}^T\mathbf{X}_n\mathbf{w},$$

where $y_r(n)$ is a received signal component, **h** represents a filter coefficients vector, **X** represents an input data matrix, **w** represents a coefficient vector.

31. The method of claim 14, wherein a least mean square update rule is applied to train at least one coefficient of the echo canceller filter, the least means square update rule is defined as

$$\mathbf{w}_{n+1} = \mathbf{w}_n - \mu e(n)\mathbf{X}_n{}^T\mathbf{h},$$

where **w** represents a coefficient vector, μ represents step size, $e(n)$ represents the error signal, and $\mathbf{X}_n{}^T\mathbf{h}$ represents a vector-matrix product where **X** is a Hankel matrix and **h** represents a filter coefficients vector.

32. The method of claim 31, wherein

$$e(n) = y_r(n) - \mathbf{h}^T\mathbf{X}_n\mathbf{w},$$

where $y_r(n)$ is a received signal component.

33. The method of claim 14, wherein a least mean square update rule is applied to train at least one coefficient of the echo canceller filter, the least means square update rule is defined as

$$\mathbf{w}_{n+1} = \mathbf{w}_n - \mu e(n)\mathbf{x}_{n-d},$$

where **w** represents a coefficient vector, μ represents step size, $e(n)$ represents the error signal, and **x** represents a data vector.

34. The method of claim 33, wherein

$$e(n) = y_r(n) - \mathbf{h}^T\mathbf{X}_n\mathbf{w},$$

where $y_r(n)$ is a received signal component, **h** represents a filter coefficients vector, **X** represents an input data matrix, **w** represents a coefficient vector.